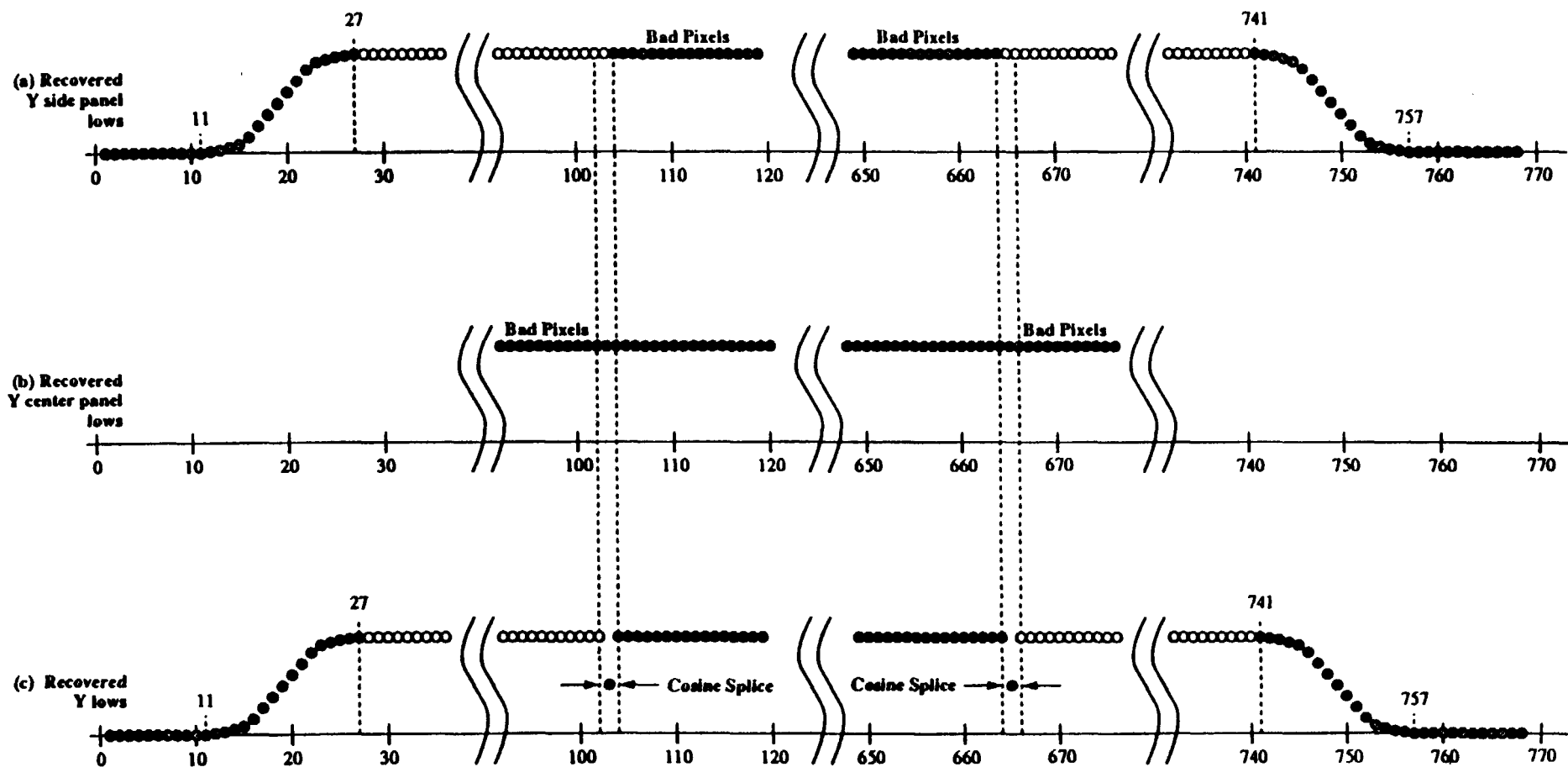
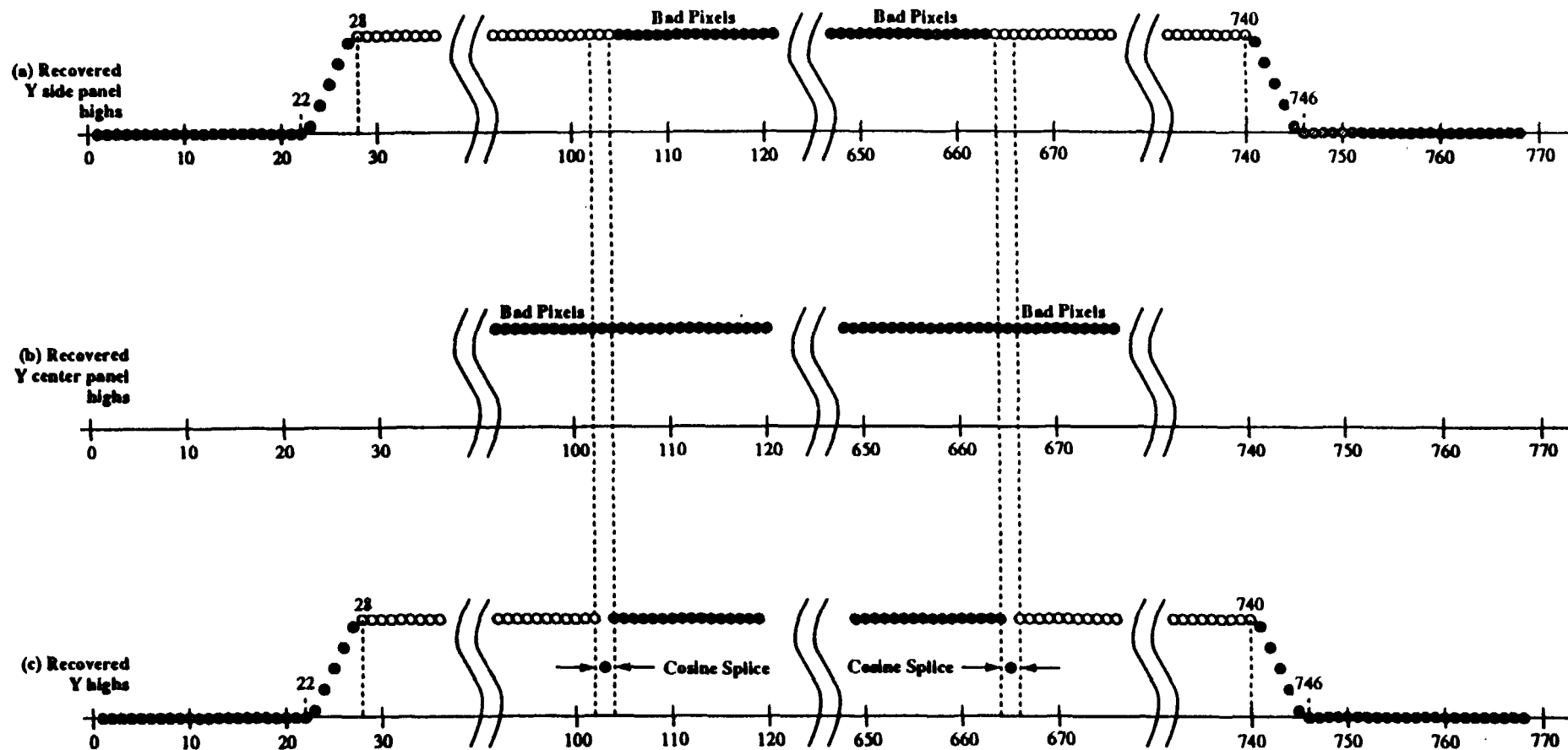


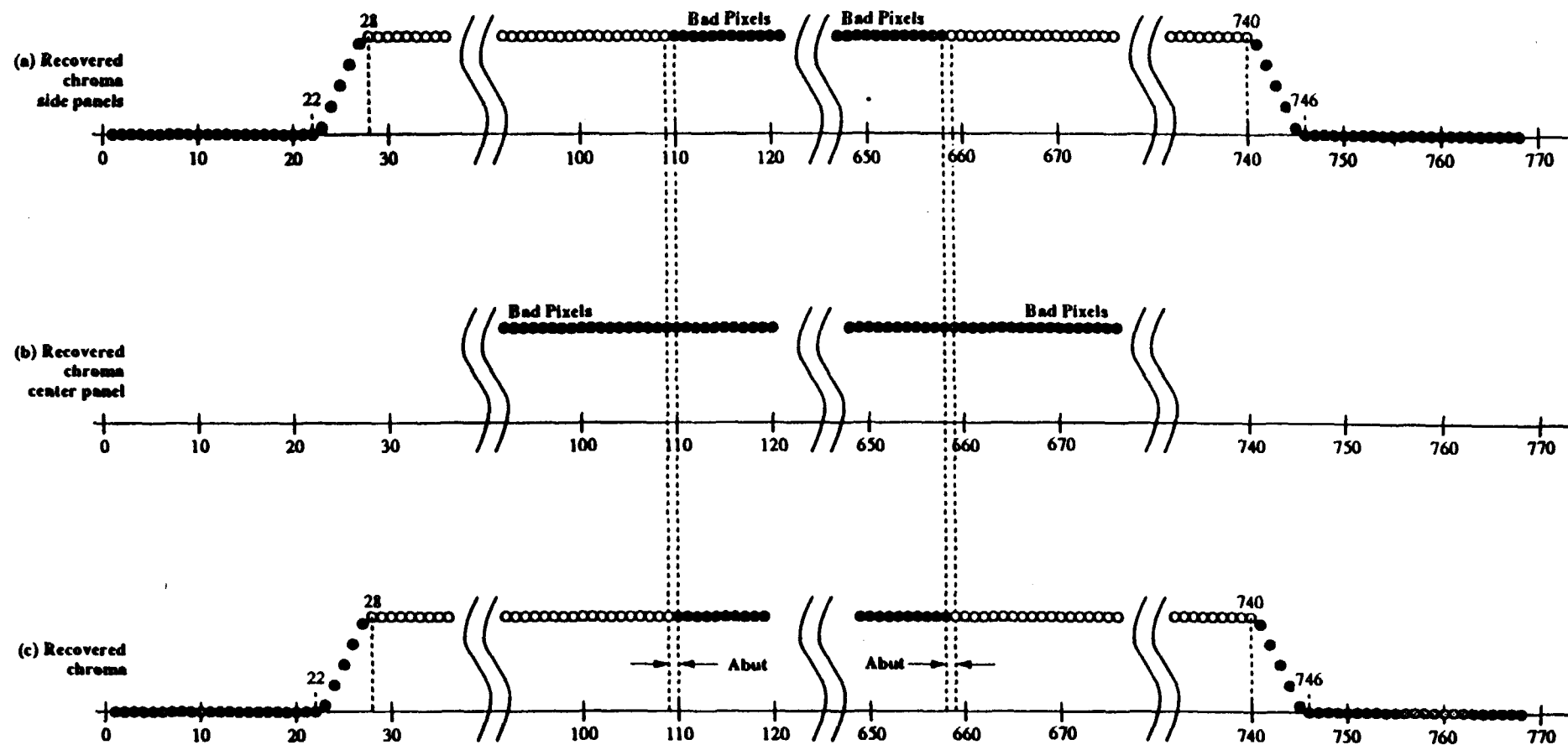
**Fig. 14 Components 1 & 2 Guardbands**



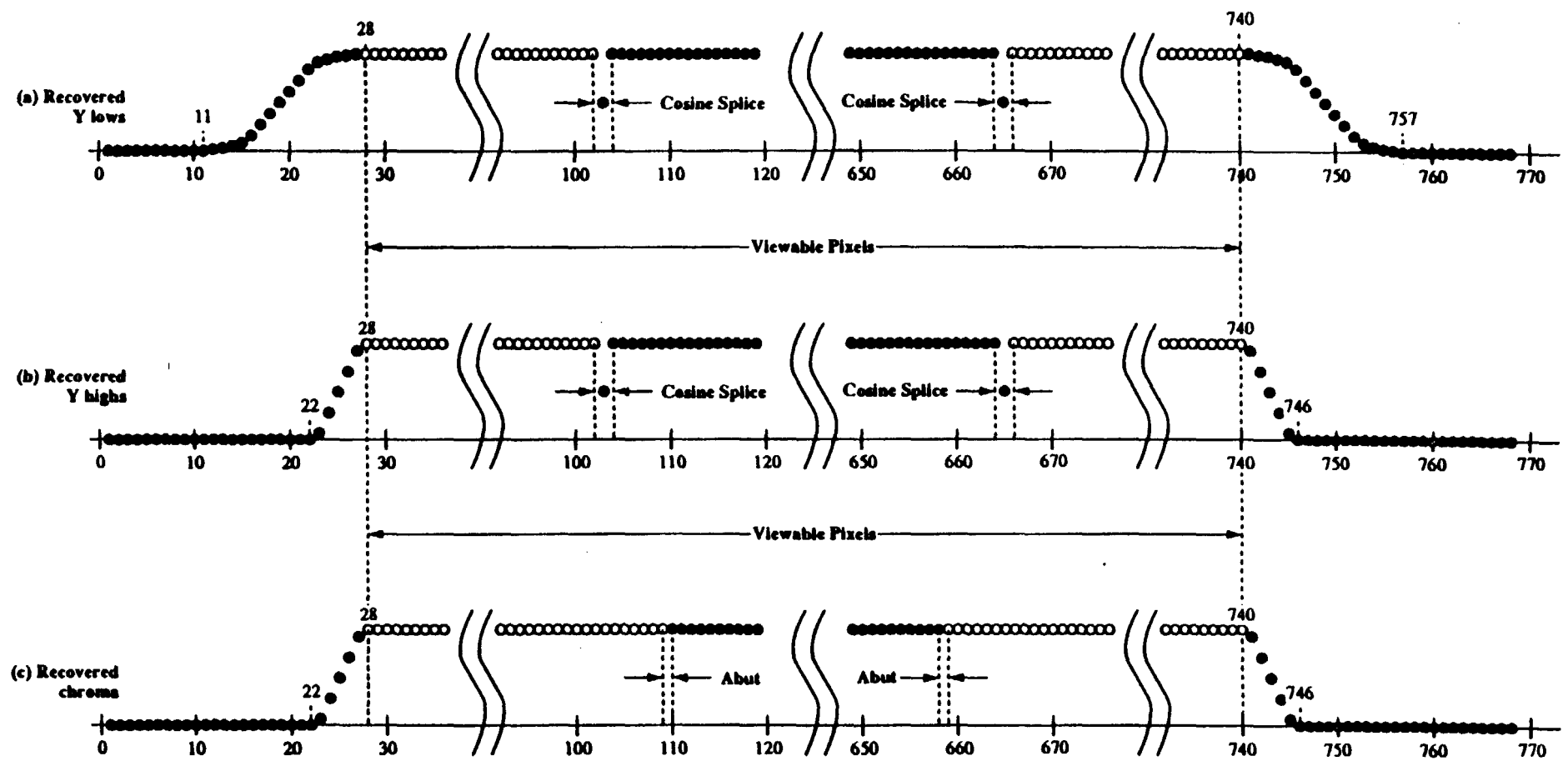
**Fig. 15 Decoder Splicing - Luma Lows**



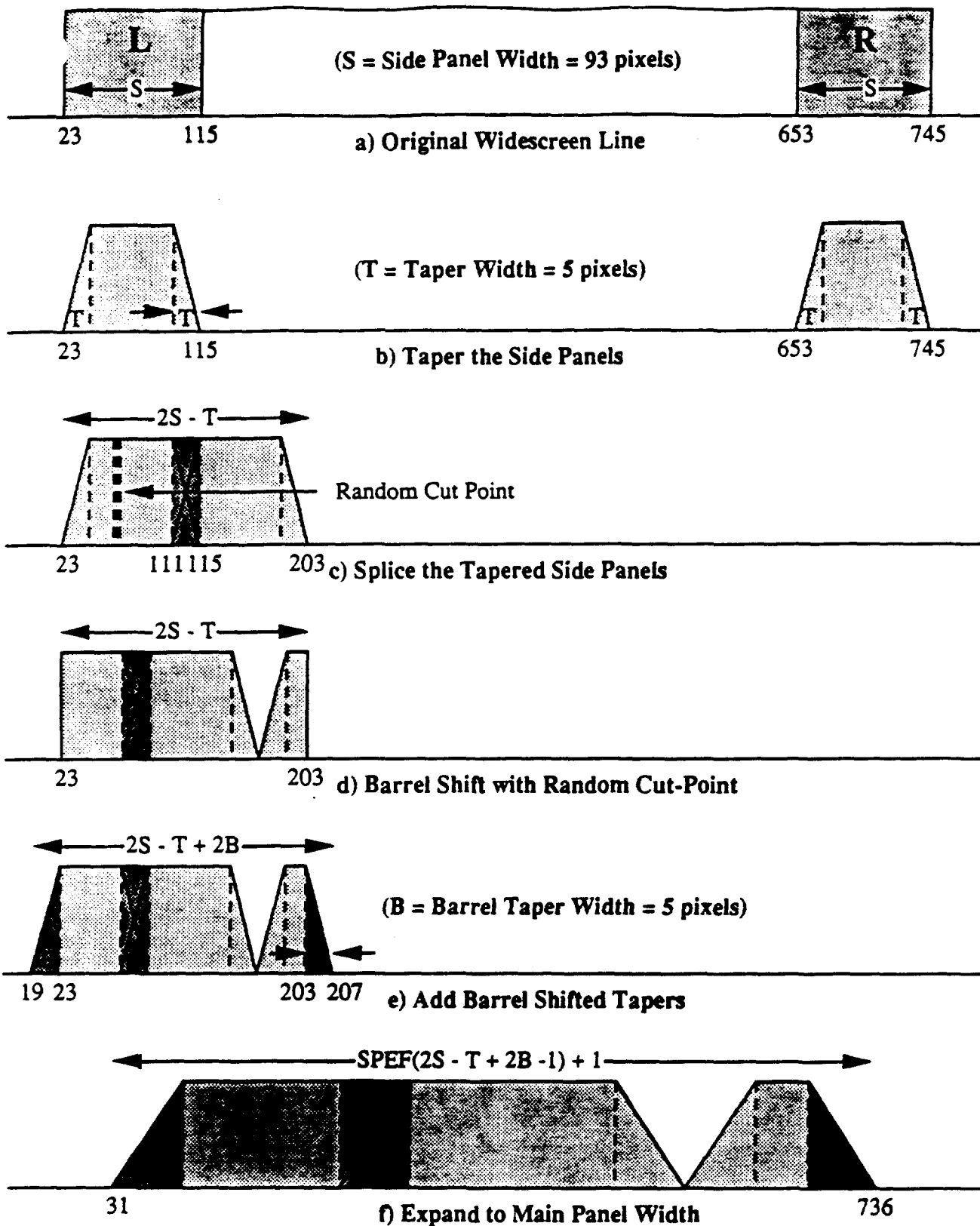
**Fig. 16 Decoder Splicing - Luma Highs**



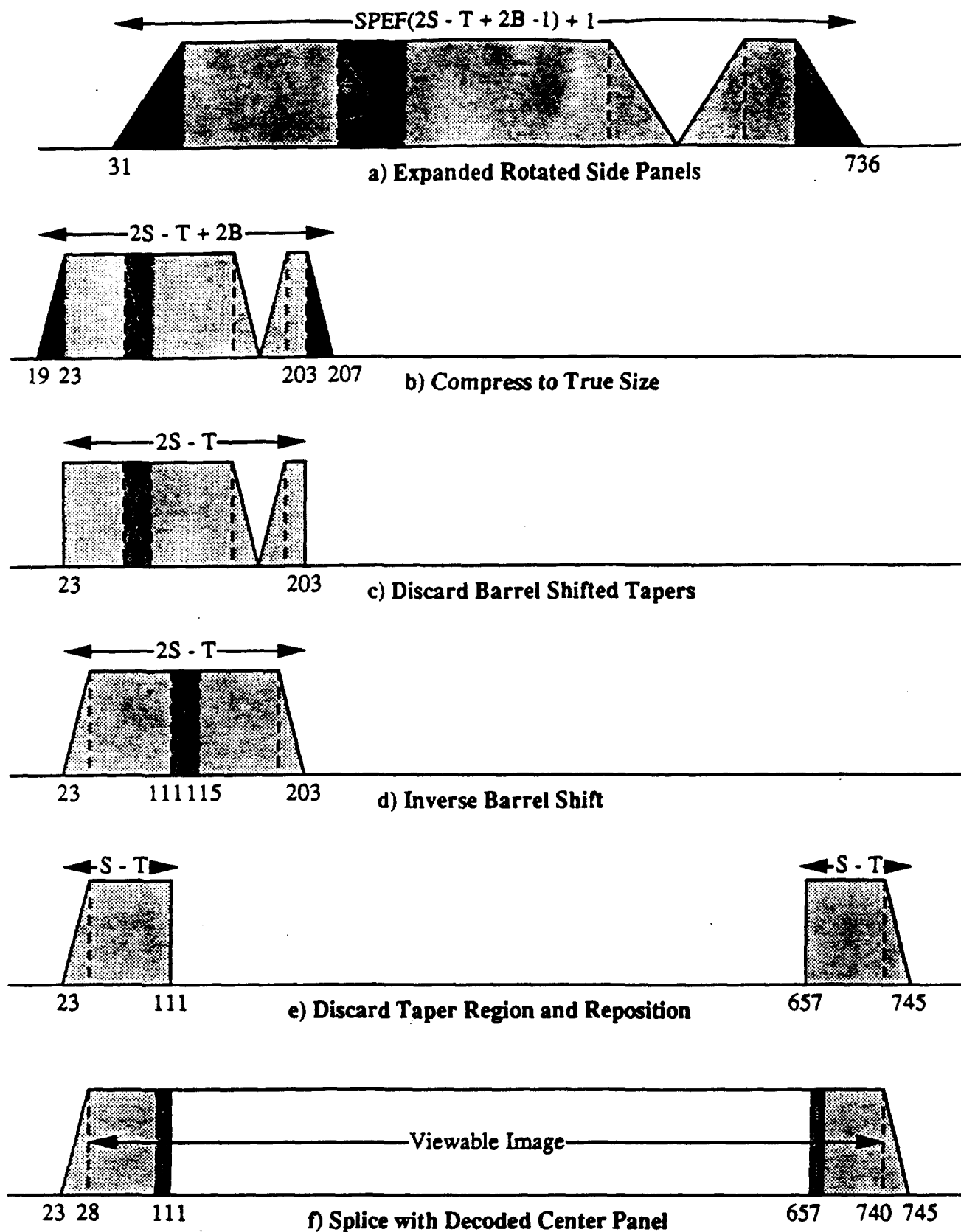
**Fig. 17 Decoder Splicing - Chroma**



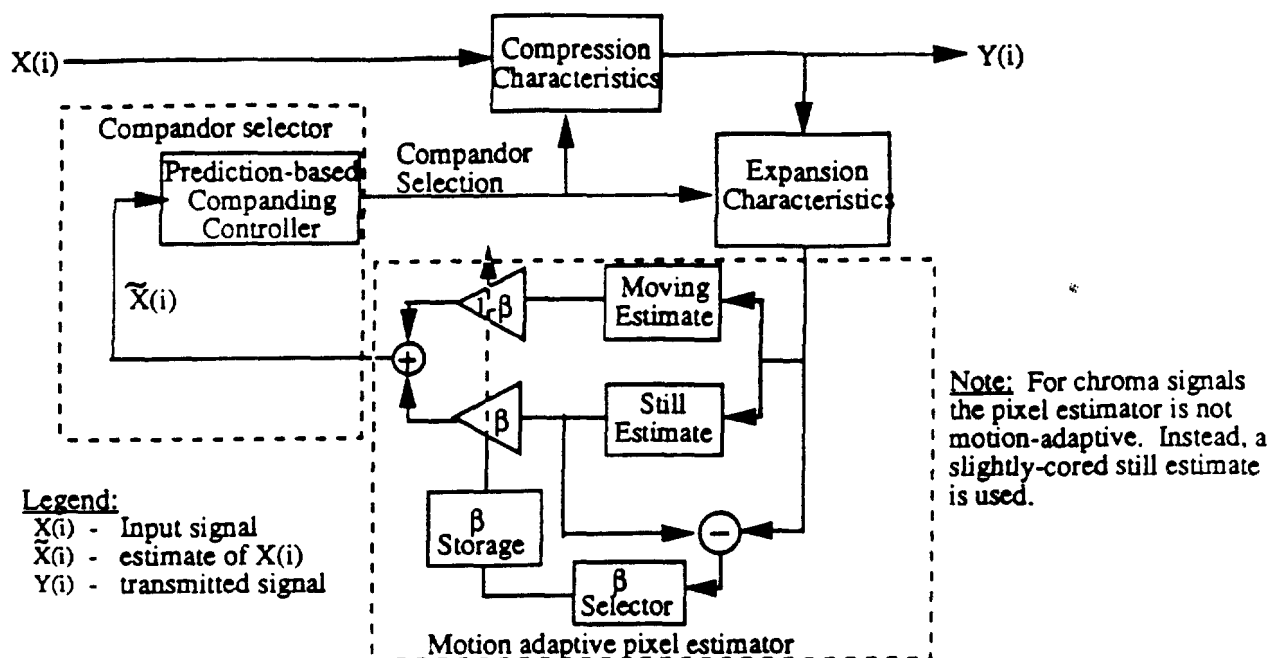
**Fig. 18 Decoder Splicing - Summary**



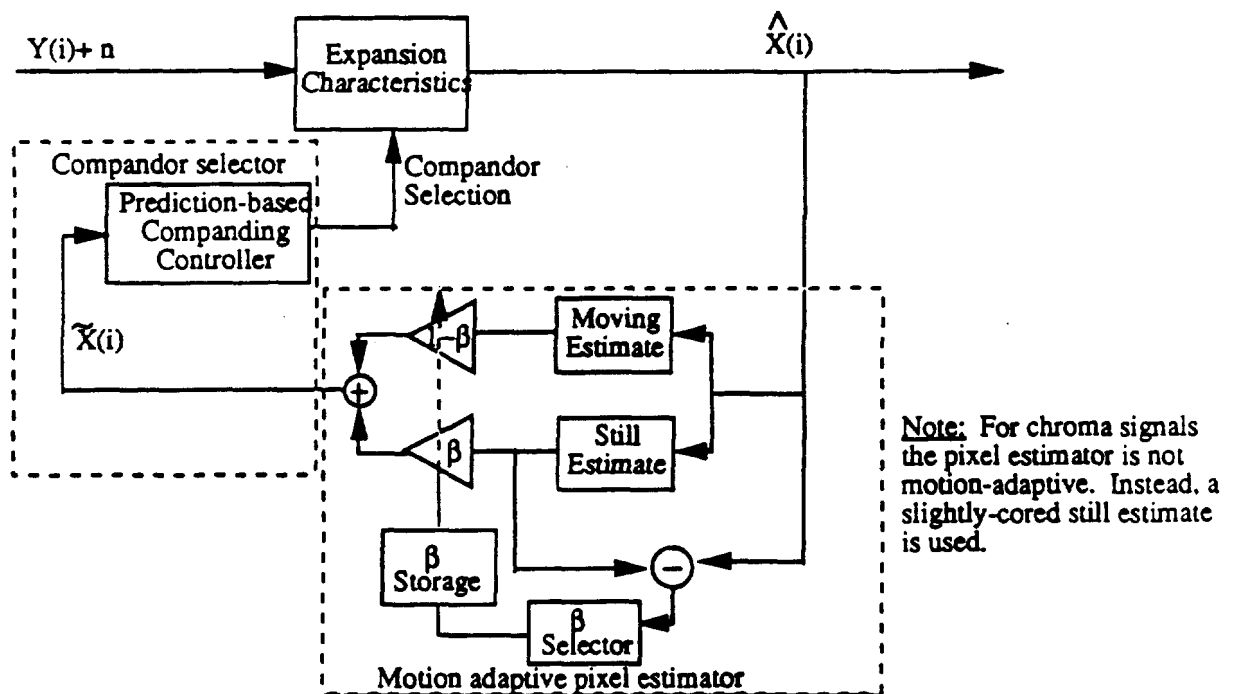
**Fig. 19 Line Rotation Using Tapers (Encoder)**



**Fig. 20 Line Rotation Using Tapers  
(Decoder)**



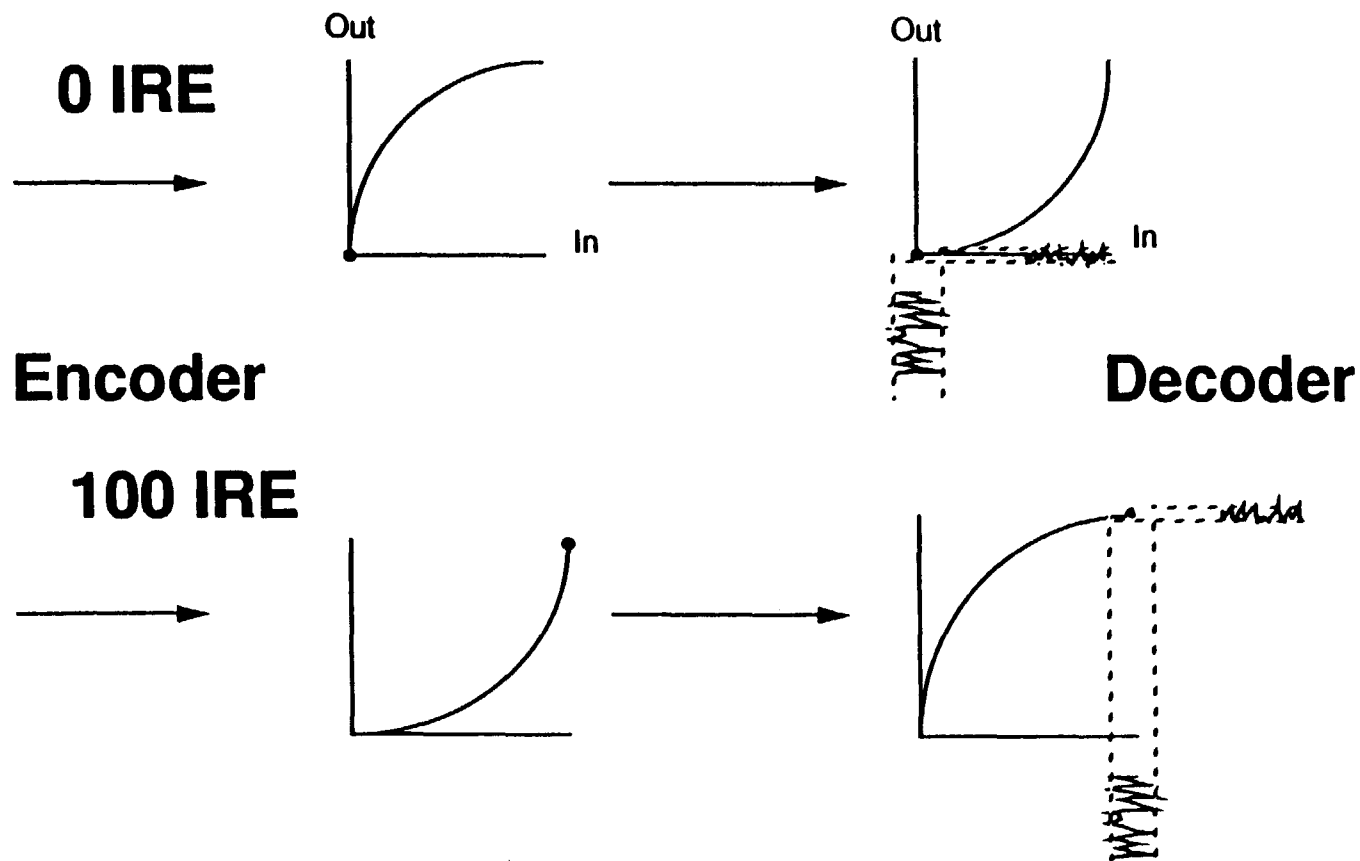
(a) Transmitter processing.



(b) Receiver processing.

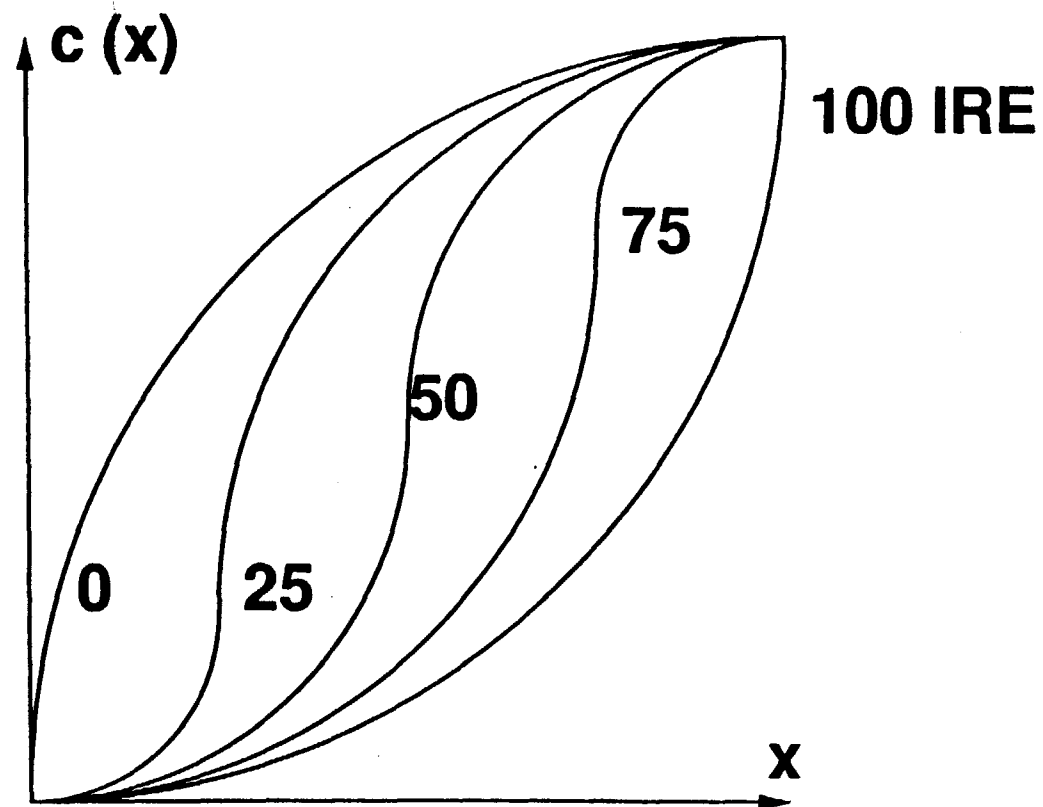
**Fig. 21 Noise Reduction Processing**

# TAILOR THE COMPANDER TO THE SIGNAL: ADAPTIVE DIRECT COMPANDING



**Fig. 22**

## A FAMILY OF COMPRESSION CURVES



**Fig. 23**

## COMPENSATED ADAPTIVE DIRECT COMPANDING SYSTEM

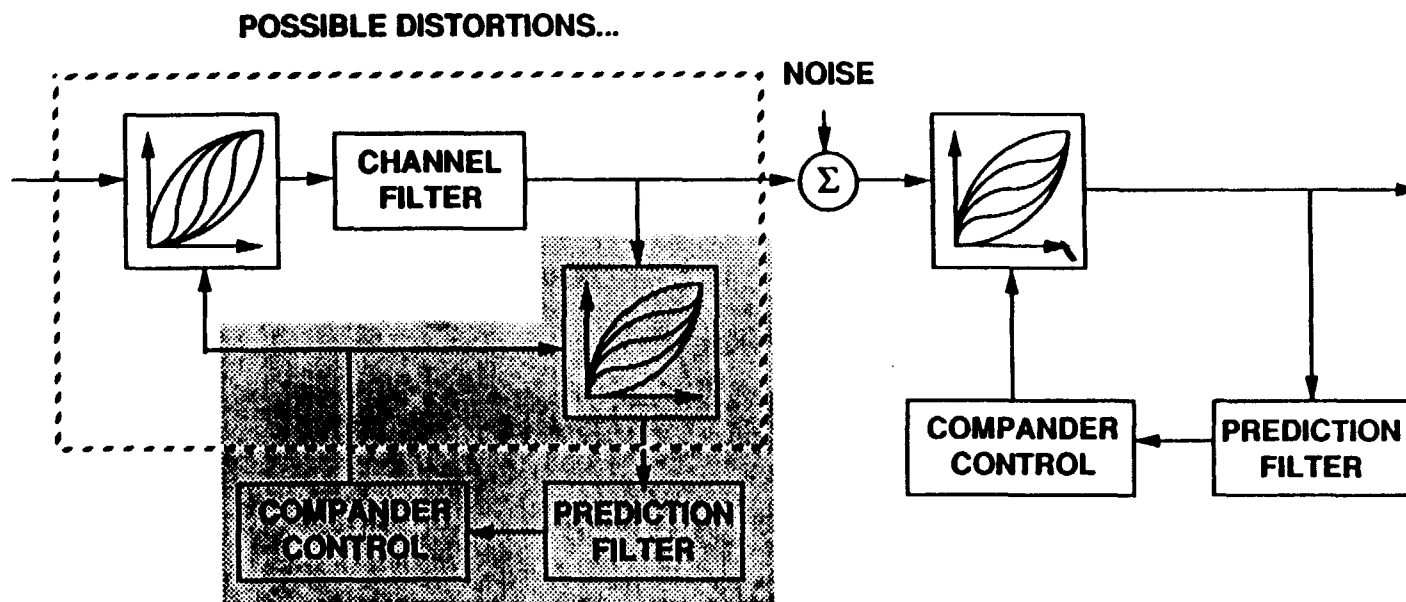


Fig. 24

# EXPANDER FUNCTION AND NOISE PDF'S

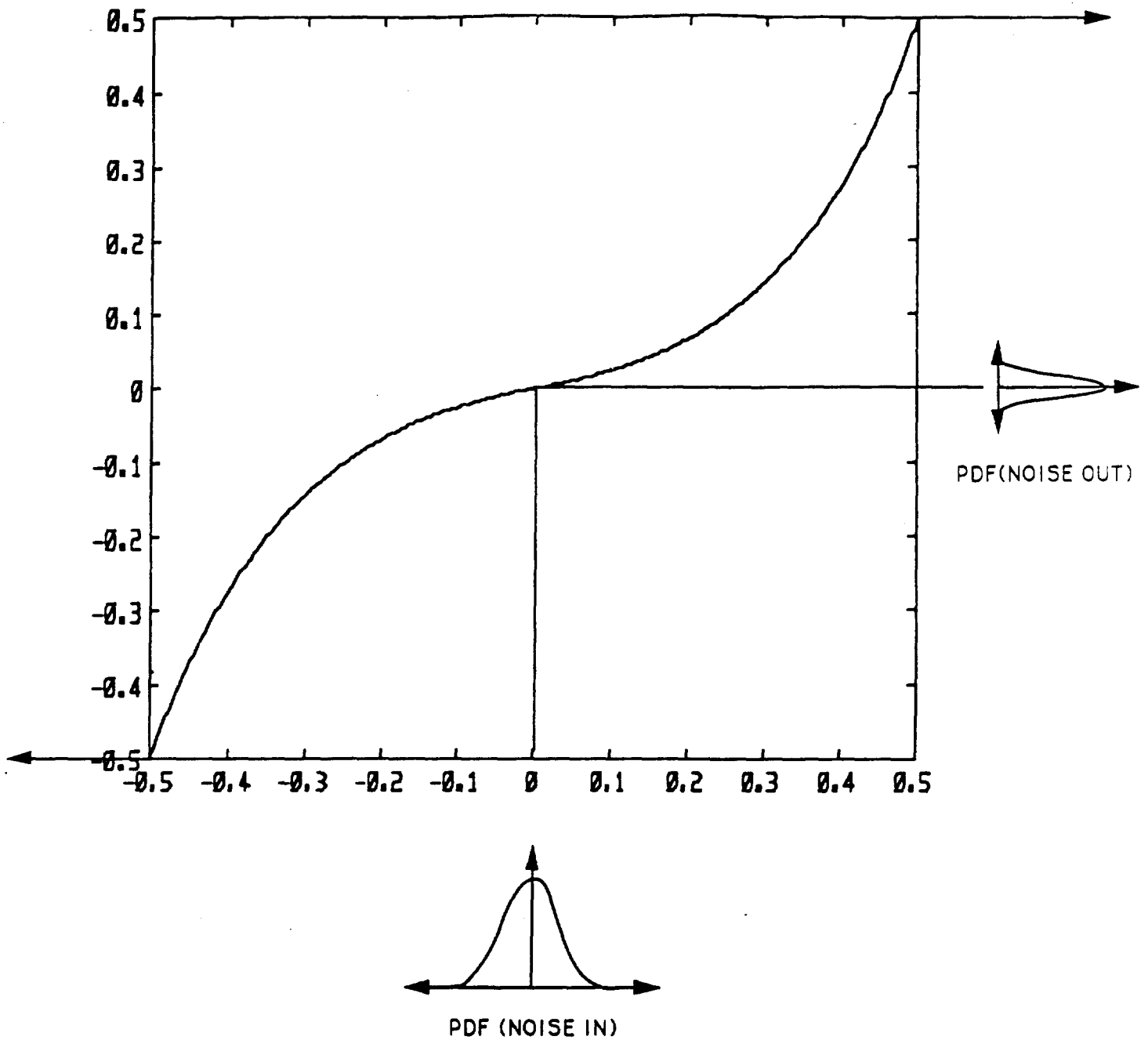


Fig. 25

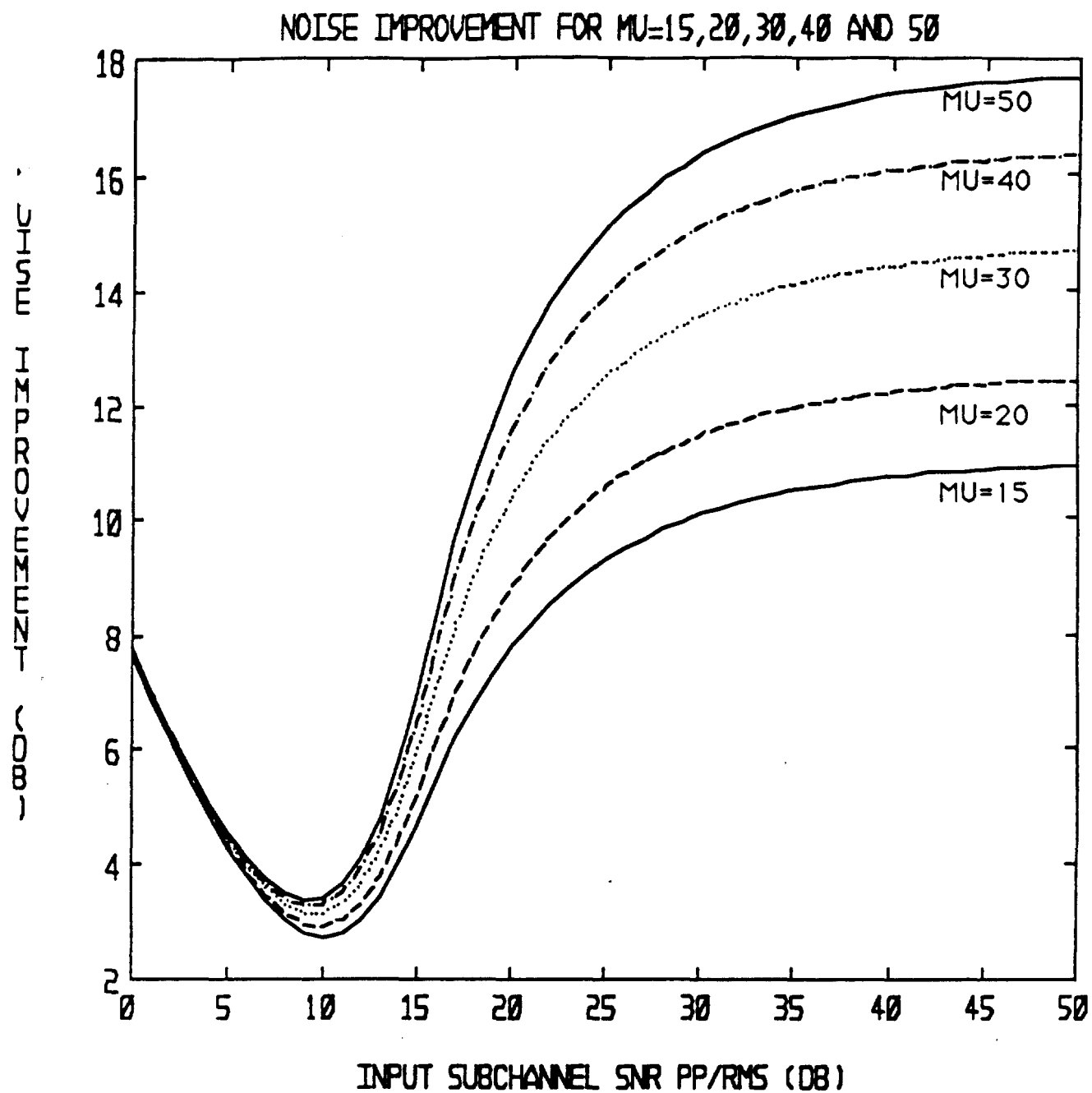
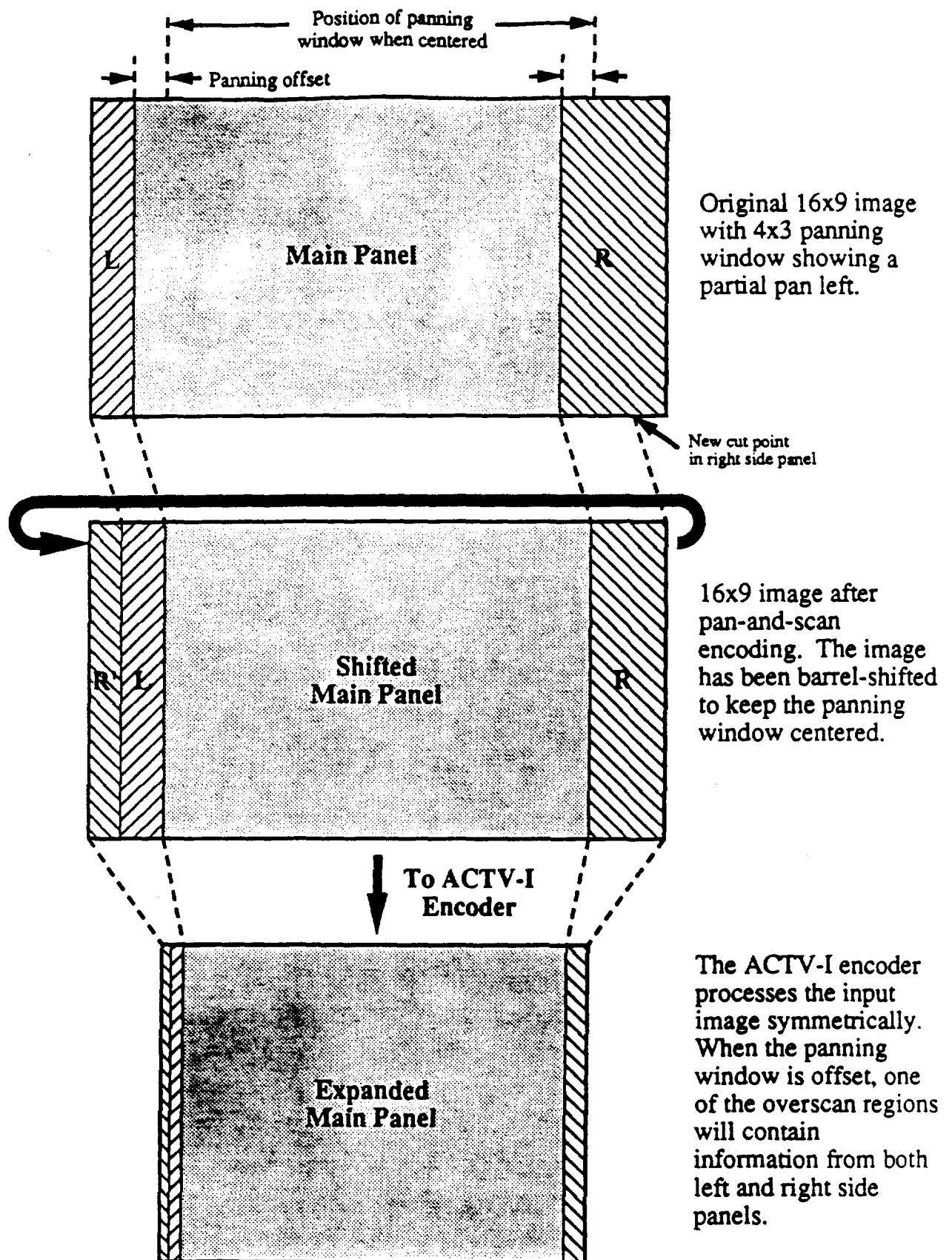
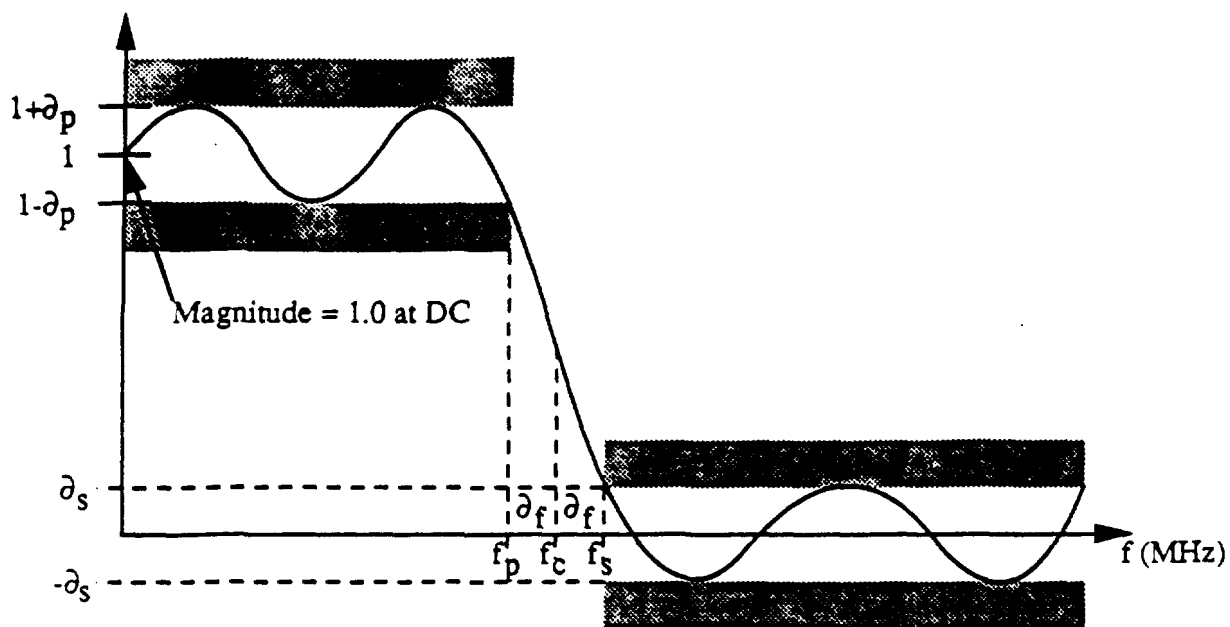


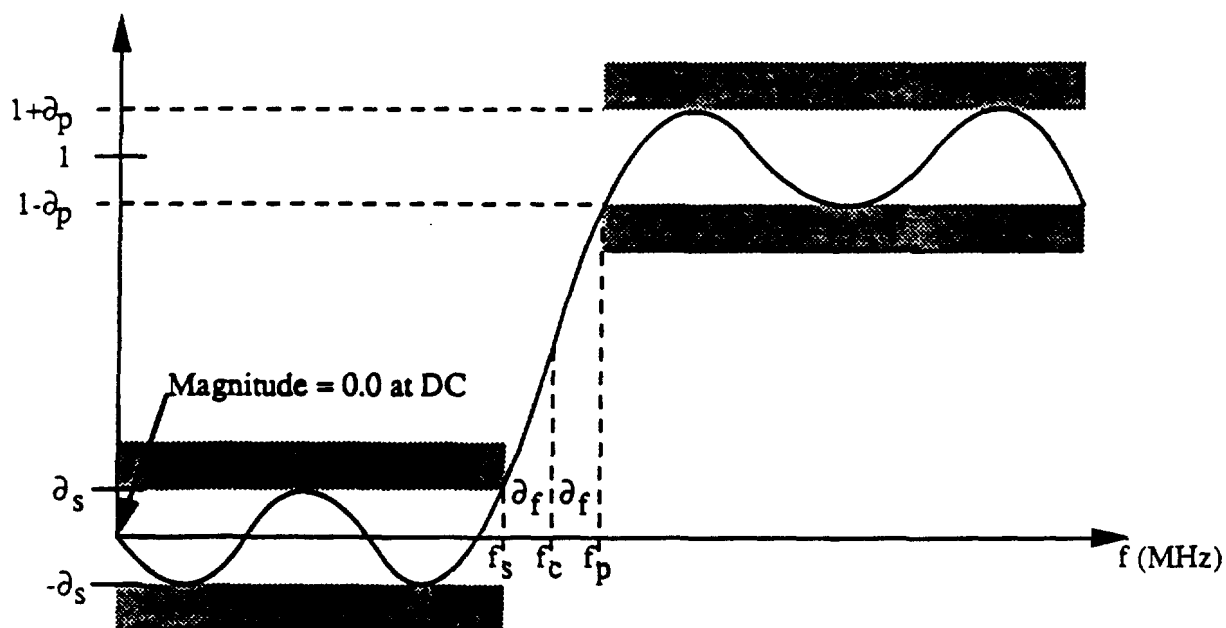
Fig. 26



**Fig. 27 Pan-and-scan implementation**



**Fig. 28 Error Tolerance for Lowpass Filter Design**



### Fig. 29 Error Tolerance for Highpass Filter Design

## Appendix A: Filter Specifications

Figures 4 and 5 show the various system filters. The function of each filter is described in words below. Figures 28 and 29 show the general format for specifying passband, stopband, transition band, and cut-off frequency; the specifications for each filter are in the accompanying Table 1.

### Encoder Filters

HZF #E1	Initial bandlimiting filter for center panel I (IC).
HZF #E2	Initial bandlimiting filter for center panel Q (QC).
HZF #E3	Bandsplit filter for center panel Y lows (YCL). This filter determines how much center panel luma will contain full V-T resolution. For a flat center panel luma response, this filter should be the complement of HZF #E5.
HZF #E4	Bandsplit filter for side panel Y lows (YSL). It determines how much side panel luma will contain full V-T resolution. Its cut-off frequency must be low enough so that the signal, after time compression, will pass undistorted through the channel filter. For a flat side panel luma response, the convolution of HZF #E4 and HZF #D9 should be the complement of the convolution of HZF #E7 and HZF #D10.
HZF #E5	Bandsplit filter for center panel Y highs (YCH). This filter determines how much center panel luma will contain half V-T resolution (be intraframe averaged). For a flat center panel luma response, this filter should be the complement of HZF #E3.
HZF #E6	Bandlimiting filter for side panel luma, without YHH (Component 3). For spectrum efficiency, this filter should be sharp. Any ringing will be canceled when YHH is added back in. Bandlimiting for center panel luma is performed by the channel, which is a very sharp filter. This gives the compatible set a sharp main picture. To yield a flat side panel response, this filter should be the complement of HZF #E10.
HZF #E7	Bandsplit filter for side panel Y highs (YSH). It determines how much side panel luma will be intraframe averaged, and it also affects crosstalk between YSH and QS.
HZF #E8	Initial bandlimiting filter for side panel Q (QS). It affects crosstalk between QS and YSH.
HZF #E9	Initial bandlimiting filter for side panel I (IS). This filter's cutoff frequency is the same as that of QS, or 0.625 MHz.

HZF #E10	Bandsplit filter for widescreen Y high-highs (YHH, or Component 3). This filter determines the lower end of the YHH band; HZF #E14 determines the upper end.
HZF #E11	This filter determines the lower limit of the band of luma frequencies prefiltered by Qbert.
HZF #E12	This filter suppresses any harmonics and repeat spectra that may be present after noise reduction and time expansion of the combined YSH/QS signal (YQS). It prevents crosstalk from modulated YQS into center panel luma.
HZF #E13	Inverse Nyquist-slope filter for modulated IS. The gain of this filter should be exactly 0.5 at 3.58 MHz
HZF #E14	This filter determines the upper band edge for YHH. The nominal upper cutoff frequency for YHH is 3.58 MHz higher than the cutoff frequency of this filter.
HZF #E15	This filter suppresses any repeat spectra that may be present after time expansion of YHH. It limits crosstalk from YHH into the ACTV signal.
HZF #E16	This filter shapes the upper band edge of the YSD signal. It should be the complement of HZF #E5.
HZF #E17	This filter shapes the lower band edge of the YSD signal. It should be the complement of HZF #E4.

#### Decoder Filters

HZF #D1	Highpass filter for modulated Component 2 plus center panel of Component 1. The complement is generated by a subtracter module. It determines the range of center panel luma frequencies that will be intraframe averaged.
HZF #D2	This filter determines the lower limit of the band of luma and modulated chroma frequencies processed by Qbert decoding.
HZF #D3	Demodulation filter for recovered IC (before raster mapping).
HZF #D4	Demodulation filter for recovered QC (before raster mapping).
HZF #D5	Bandshaping filter for center panel luma. This filter yields the complementary slope for YHH to yield a flat response.
HZF #D6	Nyquist-slope filter for modulated YQS. The convolution of this filter with HZF #E13 should be exactly double sideband around 3.58 MHz. The gain of this filter should be exactly 0.5 at 3.58 MHz
HZF #D7	Demodulation filter for recovered YQS (before raster mapping).

HZF #D8	Demodulation filter for recovered IS (before raster mapping). To prevent quadrature crosstalk from YQS, the cutoff frequency of this filter must not be larger than about 1 MHz.
HZF #D9	Final bandshaping filter for YSL.
HZF #D10	Bandshaping and separation filter for YSH. It affects flatness of side panel luma response and crosstalk from QS into YSH.
HZF #D11	Separation filter for QS. It affects crosstalk from YSH into QS.
HZF #D12	Final bandlimit filter for I. Tends to smooth out any seam artifacts.
HZF #D13	Final bandlimit filter for Q. Tends to smooth out any seam artifacts.
HZF #D14	Lower sideband reject for YHH. Should be flat above 4 MHz to insure correct bandshape for frequency seaming.
HZF #D15	Initial bandlimiting filter for Component 3. Rejects any remaining Component 1.
HZF #D16	Bandsplit filter to separate IS from YSD.
HZF #D17	Bandsplit filter to separate YSD from IS.
HZF #D18	Bandsplit filter for widescreen luma. Thus separates the full V-T lows from the half V-T (intraframe averaged) highs. This separation is needed because the highs and lows are processed separately in the progressive scan converters.

Table 1: Horizontal Filter Specifications

Filter #	Type	$f_c$ (MHz)	$\Delta f$ (MHz)	$\Delta p$	$\Delta s$	DC Gain
Encoder						
E1	lowpass	1.50	0.30	0.05	0.05	1.00
E2	lowpass	0.62	0.20	0.05	0.05	1.00
E3	lowpass	2.00	0.30	0.05	0.05	1.00
E4	lowpass	0.98	0.20	0.05	0.05	1.00
E5	highpass	2.00	0.30	0.05	0.05	0.00
E6	lowpass	5.00	0.30	0.05	0.05	1.00
E7	highpass	0.85	0.20	0.05	0.05	0.00
E8	lowpass	0.62	0.20	0.05	0.05	1.00
E9=E8	lowpass	0.62	0.20	0.05	0.05	1.00
E10	highpass	5.00	0.30	0.05	0.05	0.00
E11	lowpass	3.00	0.30	0.05	0.05	1.00
E12	lowpass	2.00	0.30	0.05	0.05	1.00
E13	highpass	3.58	0.50	0.05	0.01	0.00
E14	lowpass	2.80	0.30	0.05	0.05	2.00
E15	lowpass	1.50	0.30	0.05	0.05	4.00
E16=E3	lowpass	2.00	0.30	0.05	0.05	1.00
E17	highpass	0.85	0.20	0.05	0.05	0.00
Decoder						
D1	highpass	1.80	0.30	0.05	0.05	0.00
D2	lowpass	2.50	0.30	0.05	0.05	1.00
D3	lowpass	2.00	0.30	0.05	0.05	2.00
D4	lowpass	0.62	0.40	0.05	0.05	2.00
D5=E6	lowpass	5.00	0.30	0.05	0.05	1.00
D6	lowpass	3.58	0.50	0.05	0.05	1.00
D7	lowpass	2.00	0.30	0.05	0.05	2.38
D8	lowpass	0.62	0.30	0.05	0.05	2.86
D9=E4	lowpass	0.98	0.20	0.05	0.05	1.00
D10=E7	highpass	0.85	0.20	0.05	0.05	0.00
D11	lowpass	0.62	0.20	0.05	0.05	1.60
D12	lowpass	1.50	0.70	0.05	0.05	1.00
D13	lowpass	1.00	0.32	0.05	0.05	1.00
D14	highpass	4.30	0.40	0.05	0.05	0.00
D15=E15	lowpass	1.50	0.30	0.05	0.05	4.00
D16=E8	lowpass	0.62	0.20	0.05	0.05	1.00
D17=E7	highpass	0.85	0.20	0.05	0.05	0.00
D18=E3	lowpass	2.00	0.30	0.05	0.05	1.00

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# DESCRIPTION OF THE DIGITAL AUDIO SYSTEM FOR ACTV

## 1. Introduction

It was stated in previous descriptions of Advanced Compatible Television systems [1] that digital audio will be transmitted as a data carrier in the lower VSB portion of the RF spectrum, the center frequency of which is about 1.1 MHz from the picture carrier frequency. Figs. 1 and 2 show the principles of such a data transmission.

Exceptionally good quality digital audio compression methods are beginning to be available, requiring 256 KBPS data rate for a stereo channel (two mono channels). The ACTV system will adopt an audio system at this data rate. The bandwidth of the data carrier, after the inclusion of Forward Error Correction (FEC) will be no more than 220 KHz.

The Motion Picture Engineering Group (MPEG) of the International Standards Organization (ISO) has classified digital audio compression methods into four categories (clusters), in the context of selecting a standard for a variety of applications, including digital over-the-air broadcast. These are:

1. Transform Coding with Overlapping blocks (also referred to as Transform Coding with Time Domain Alias Cancellation)
2. Subband Coding with less than 8 subbands
3. Subband Coding with more than 8 subbands
4. Transform Coding with Nonoverlapping Blocks

The Transform Coding can potentially deliver the best quality audio. This is because the Transform method has the best ability to match the critical band structure of the human ear. Within such methods, coding with overlapping blocks and time domain alias cancellation has the potential of giving better reconstructability and frequency response. Such a method is incorporated in a compression method referred to as ASPEC (Adaptive Spectro-Perceptual Entropy Coding). This system is being jointly developed by the following organizations:

American Telephone and Telegraph, Murray Hill, NJ, USA  
CNET (France Telecom), Rennes, France  
Fraunhofer Gesellschaft, Erlangen, West Germany  
Deutsche Thomson-Brandt GmbH, Hannover, West Germany.

In the following sections principles of the ASPEC compression system and methods of over-the-air delivery will be described.

*The hardware to be provided to the ATTC will consist of the ASPEC system, which is predicated under the assumption that it will be the chosen standard by the MPEG. In the event that one of the other systems (at the data rate of 256 KBPS) will be chosen by the MPEG, consideration will be given to making that as the ACTV digital audio standard. From the transmission point of view all well designed 256 KBPS audio systems will have essentially the same quality attributes, i.e., they will exhibit high audio quality at a bit error rate of better than  $10^{-4}$ , and will begin to fail at an error rate of about  $10^{-3}$ . The data modulation will be QPSK, irrespective of which standard is chosen, and will occupy a bandwidth of about 220 KHz.*

## **2. Critical Bands of the Human Ear**

The fundamental premise of the compression schemes listed in the above section is that complex audio signals, which require the standards such as Compact Disk, contain at any given time, much information that the human ear cannot hear. The characteristics of the human ear are known almost entirely in terms of the frequency domain. Thus, it is in the frequency domain that parts of the audio signal which can either be discarded altogether or transmitted less precisely, with psychoacoustically benign effect, can be identified.

The characteristics of the human ear on which the above compression schemes depend is the *Masked Threshold of Hearing*. If there are two signals, one strong and the other weak, simultaneously present, the weak signal is inaudible to the ear if the sound pressure of the weak signal is below the masked threshold of hearing. This threshold power depends on the spectra and power of the strong signal. In particular, if the weak signal is quantization noise, it can be made inaudible (i.e., the number of bits producing the quantization noise can be selected to make it inaudible), provided it is lower than the masking thresholds that are determined by the strong signal.

*Critical bands* are those frequency bands of the human ear which are such that, if the masking and the masked signals are within such a band, then the masking phenomenon is maximized. Hence division of the broadband auditory spectra into critical bands, and carefully matching the signal and quantization noise levels in each band, will result in errors that are psycho-acoustically most benign.

Table 1 and Figure 3 show the critical band parameters; these are based on reference [2].

### 3. Masking Thresholds

The masked thresholds of hearing are made up of two components. The first of them is the *Absolute Threshold of Hearing* (see Fig 4). This is simply the lowest sound pressure level at which a pure sine wave is audible as a function of frequency, in the absence of any other acoustic stimuli. Suppose that, say, a narrowband noise ("masking signal") of 1 KHz is present in the audio channel. This raises the threshold of hearing to different shaped curves, that depend on the power of the masking signal. A signal ("masked signal") below these will be made inaudible by the presence of the masking signal. Such data is available in references 3 to 5. Given the short-term spectral distribution of the input signal, and the boundaries of the critical bands, the digital audio encoder computes the masking thresholds in each band (see section 5). This information is used to assign the bits for transmission.

### 4. Transformation of Input Time-Samples into the Frequency Domain

#### *General:*

The input signal is time sampled into blocks and converted to a spectral representation. This is accomplished via a "*Modified Discrete Cosine Transform*" (MDCT) using the "*Time Domain Alias Cancellation*" (TDAC) method as described in reference 6. Figures 5 and 6 show the encoder and decoder block diagrams.

One data block to be transformed contains 1024 time samples, i.e. at a sampling frequency of 48 KHz this block contains the data of 21.33 ms of sound. The consecutive data blocks overlap by 50% and therefore the true "*block pitch*", i.e. the time between consecutive transform operations,